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METHOD AND APPARATUS FOR ESTABLISHING A VOICE CALL
TO A PSTN EXTENSION FOR A NETWORKED CLIENT COMPUTER

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to the field of telecommunications and, in particular, to a method and apparatus for establishing a voice call to a Public Switching Telephone Network (PSTN) extension for a networked client computer.

2. Background Information

Numerous advances have been made in recent years in the field of telecommunications. In particular, the field of internet telephony has emerged as a viable technology that is evolving at an ever increasing rate. Evidence of this evolution of internet telephony is best characterized by the number of new products recently become available in the market. Products such as CoolTalk by Netscape Communications Corporation of Mountain View, California; Internet Connection Phone by International Business Machines of Armonk, New York; Intel Internet Phone (IPhone) by Intel Corporation of Santa Clara, California; NetMeeting by Microsoft Corporation, Redmond, Washington; Quarterdeck WebTalk by Quarterdeck Corporation of Marina Del Rey, California; TeleVox by Voxware Incorporated of Princeton, New Jersey; and WebPhone by Netspeak Corporation of Boca

Raton, Florida, are representative of the current state of applications facilitating internet telephony.

Each of these products offers internet based voice communications with a telephone motif, between two users each using the same (or compatible) product on either end of the internet connection. That is, the internet provides the "switching" architecture for the system, while the computer acts as the "handset", or the audio interface. One reason for the proliferation of these applications is a desire to push the technology of the internet to provide a total communications tool. The appeal to users is that, currently, the use of the internet is free of toll charges. Therefore, currently, a user of an internet phone product may communicate with another user located anywhere else in the world without having to pay the long distance charges associated with making a telephone call using the public switched telephone network (PSTN).

However, consumers expecting to completely eliminate their long-distance telephone bills through the exclusive use of internet telephony are in for a disappointment. As they shall soon discover, although innovative in their own right, the current internet based telephony applications identified above have a number of limitations which retard their acceptance as a primary communications tool. One such limitation is that many of the applications identified above require that both users have installed the same software package, or compatible packages and, therefore, provide a relatively low level of interoperability. One reason for this lack of interoperability between internet telephony applications is that the developers of many of these products have incorporated different

voice encoders (commonly referred to as a "voice codec", or simply a "codec" by those in the telecommunication arts) in the products. Consequently, as a result of the different codecs used, many internet telephony applications are unable to recognize speech encoded (i.e., digitized) by a codec of a dissimilar application.

This problem is alleviated for those products that are upgraded to comply with emerging telephony standards, such as International Telecommunication Union's (ITU) H.323 standard. However, other limitations remain. For example, another limitation associated with many of these products is that they are tied to the internet, often requiring all users to access a common server in order to maintain a directory of available users in which to call. That is to say, many of the applications identified above do not integrate the packet switched network of the internet with the circuit switched public switched telephone network (PSTN). Therefore, although a computer connected to the internet may communicate with another user on the internet, assuming they are both using a common software application (or at least applications with compatible codecs), these applications do not support communication with a user of a Telephone handset.

The reason for this limitation is readily understood by those who appreciate the complexity of the two networks. As alluded to above, the internet is a packet switched network. That is to say, communication over the internet is accomplished by "breaking" the transmitted data into varying-sized packages (or "packets"), based primarily on communication content, and interleaving the various-sized packages to best utilize the bandwidth available at any given time on the internet. When the packets reach their intended

destination, they must be reassembled into the originally transmitted data. Loss of packets, and thus data, occur frequently in such a network, and the ability of the network to successfully transmit information from one point in the network to another determines the quality of the network. For inter-computer communication transactions involving non real-time data, the ability to transmit packets and retransmit any packets that are perceived to have been dropped is not a severe limitation and may not even be perceived by the user of the system. However, in a voice communication transaction, the delay required to retransmit even one data packet may be perceived by a user. At best, such delays are an annoying inconvenience. In practice, the delays actually can become intolerable, as the cumulative latency adds up with successive transmissions.

In contrast to the packet switched network of the internet, the public switched telephone network (PSTN) is a circuit switched network. That is to say that the PSTN assigns a dedicated communication line to a user with which to complete the telephone call, wherein the user can utilize the assigned resource of the PSTN in any way they choose, with the understanding that the user is paying for the use of the dedicated resource of the PSTN. While the circuit switched approach of the PSTN system is not necessarily the most efficient system in terms of call traffic (i.e., it does not make use of the "dead space" common in a conversation), it is relatively easy to ensure that information destined for a particular user is delivered, it simply provides a dedicated line to complete the transaction.

Nonetheless, despite these engineering challenges, a few products have emerged which purport to integrate the PSTN to the internet. Products such as Net2Phone by IDT

Corporation of Hackensack, New Jersey, claim to provide a computer user with the ability to place and receive a phone call to/from a PSTN extension. Unfortunately, none of these products completely solve the problem of integrating the two networks. The limitations perhaps best characterized by way of an example communication session. With these prior art internet telephony applications, a user of an internet telephony enabled client computer initiating a telephone call to a Telephone handset launches the voice call from the client computer by accessing a server (the primary access server), operated by the developer of the internet telephony application that supports internet telecommunications. As the initiator accesses the primary access server, he/she is prompted for a destination address, which takes the form of a PSTN telephone number for an outgoing call to a Telephone handset. Having provided the primary access server with the PSTN telephone number associated with the Telephone handset, the primary server somehow determines¹ which server in a community of similarly enabled servers (i.e., servers with the hardware/software necessary to provide access to the PSTN) is closest to the destination address, and completes the telephone call by routing the telephone call through a number of intermediate servers on the internet to the selected server, which will actually place the voice call to the Telephone handset on behalf of the client computer, facilitating the voice call between the client computer and the Telephone handset. In other words, the user of the client computer is required to have prior knowledge of the destination phone number, which is limiting in many circumstances. For example, in a situation where the user of the client computer is engaged in a data communication session involving a webpage for a corporate entity, the user may wish to speak with someone in a "local office" of the corporate entity. Prior art internet telephony applications require that the

¹ The manner in which the "primary access server" determines the "call originating server" is not known.

telephone number for the "local office" of the corporate entity be provided by the user of the client computer in order to place the telephone call. If the telephone number for the "local office" of the corporate entity is not provided by the webpage, the user of client computer must look it up or have prior knowledge of it.

Additionally, while the prior art approach of simply finding the internet telephony enabled server closest to the destination address may offer the simplest technical solution and a seemingly cheaper connection, it does not ensure the quality of the voice connection. One skilled in the art will appreciate that there are a number of characteristics which may impact the quality of the voice connection. For example, insofar as the internet is a packet switched network, as the number of intermediate routers required to interface the client computer to the selected server increases so, too, does the likelihood that data packets containing voice information could be lost or corrupted. The result of lost or corrupted data packets is broken or garbled speech. Another factor affecting internet telephony communication performance is the bandwidth available on the selected server. If, for example, the selected server is very busy handling a number of other processes, the performance associated with each of the processes begins to degrade (i.e., slow down), which may also result in delayed delivery of data packets containing speech, which in turn results in user perception of poor quality. Therefore, while it is important to some users of internet telephony applications to simply keep the cost down, quality considerations must also be accounted for to enable internet telephony to evolve into a viable communications tool.

Thus, a need exists for a method and apparatus for establishing a voice call to a PSTN extension for a networked client computer that is unencumbered by the limitations associated with the prior art.

SUMMARY OF THE INVENTION

In accordance with the teachings of the present invention, a method and apparatus for establishing a voice call to a PSTN extension for a networked client computer, and routing the voice call off of the network, is provided. In a first embodiment, the apparatus comprises a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for facilitating establishment of the voice call to the PSTN extension, and an execution unit, coupled to the storage medium, for executing the plurality of programming instructions. The set of communication services include services for receiving a Push-To-Talk™² event notification from the networked client computer, notifying of client computer's desire to have voice communication, and in response, determining the PSTN extension. The Push-To-Talk™ event notification includes identification information of an originating web server of the Push-To-Talk™ option, and the determination of the PSTN extension is made in accordance with the originating web server's identification information.

In another embodiment, the set of communication services further include services for providing the client computer with the IP address of the apparatus, and receiving a supplemental call request from the client computer. The supplemental call request includes the IP address, telephony and geographical information of the client computer, and the determination of the PSTN extension is also made in accordance with the telephony and/or geographical information of the client computer.

² Push-To-Talk™ is a Trademark of eFusion™, Incorporated of Beaverton, Oregon.

In yet another embodiment, the set of communication services further include services for soliciting inputs from one or more other apparatuses on one or more operating characteristics associated with establishing and facilitating the voice call between the client computer and the PSTN extension, selecting either the apparatus itself or one of the other apparatuses to complete the telephone call based on the solicited inputs, registering the voice call with the selected apparatus, and identifying the selected apparatus to the client computer.

In yet another embodiment, the set of communication services further include services for accepting registration of the voice call, receiving placement of a net phone call from the client computer, placing a PSTN phone call to the PSTN extension, and bridging the net and the PSTN phone calls. The bridging services include services for converting and delivering call signals between the client computer and the telephone handset.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be described by way of exemplary embodiments, but not limitations, illustrated in the accompanying drawings in which like references denote similar elements, and in which:

Figure 1 is a block diagram illustrating an exemplary communication system incorporating the teachings of the present invention;

Figures 2A and 2B are flow charts illustrating one embodiment of a method for establishing a voice communication session between a client computer and a PSTN handset, in accordance with one embodiment of the present invention;

Figure 2C is a flow chart illustrating one embodiment of a method for establishing a direct telephony connection between the client computer and the exemplary communication system;

Figure 3 is a block diagram illustrating an exemplary computer server incorporating the teachings of the present invention;

Figure 4 is a block diagram illustrating the software architecture of the exemplary server of Figure 3, in accordance with one embodiment of the present invention; and

Figures 5, 6 and 7 are graphical illustrations of alternate embodiments of Push-To-Talk™ indicators for requesting a voice communication session, suitable for use in the exemplary communication system described in Figures 1-4.

DETAILED DESCRIPTION

In the following description, for purposes of explanation, specific numbers, materials and configurations are set forth in order to provide a thorough understanding of the present invention. However, it will be apparent to one skilled in the art that the present invention may be practiced without the specific details. In other instances, well known features are omitted or simplified in order not to obscure the present invention. Furthermore, for ease of understanding, certain method steps are delineated as separate steps, however, these separately delineated steps should not be construed as necessarily order dependent in their performance.

Referring now to Figure 1, a block diagram is presented illustrating an exemplary communication system 100 incorporating the teachings of the present invention for connecting a user of an internet telephony enabled client computer with a user of a PSTN endpoint (e.g., telephone handset), and routing the voice call off of the internet. While the present invention will be described in the context of this exemplary computer system, based on the descriptions to follow, those skilled in the art will appreciate that the present invention is not limited to this embodiment, and may be also practiced with intranet (in lieu of the internet) and/or automated/computerized telephony answering equipment (in lieu of telephone handsets).

For the illustrated embodiment, client computer 102 incorporated with the teachings of the present invention, while in data communication with a web server, e.g. web server 128, through PSTN 140 and Internet 150, is presented with a Push-To-Talk™ option by web server 128. When client computer 102 selects the Push-To-Talk™ option, a server incorporated with the teachings of the present invention (e.g., bridgeport 162) automatically determines an appropriate destination PSTN extension, e.g. the PSTN extension of telephone handset 142, as well as an appropriate one of the community of internet/PSTN changeover servers (e.g., bridgeports 162 and 165) to place the voice call to the PSTN extension and facilitate the voice call between the user of client computer 102 and the user of a PSTN endpoint (e.g., handset 142). In the context of the example embodiment, for differentiation and ease of explanation, bridgeport 162 will be referred to as a page bridgeport, while the selected internet/PSTN changeover server (e.g., bridgeport 165) will be referred to as a changeover bridgeport. In one embodiment, the Push-To-Talk™ option is pre-associated with bridgeport 162 by web server 128, and the determination of the destination PSTN extension by bridgeport 162 is made in accordance with one or more attributes of web server 128, such as the identity of web server 128, and optionally, one or more attributes of client computer 102, such as the zip code of the area client computer 102 is located. In alternate embodiments, page bridgeport 162 is not pre-associated with the Push-To-Talk™ option, but rather is selected dynamically by web server 128.

Client computer 102, web servers 120 and 128, bridgeports 162 and 165, and handset 142 are communicatively coupled to each other by way of PSTN 140 and internet 150 as shown. More specifically, client computer 102 is coupled to internet 150 by way of a direct connection to internet service provider (ISP) 112. Client computer 102 is coupled to ISP 112 through PSTN extension 104, communication line 106 and PSTN 140. In other words, for the illustrated embodiment, client computer 102 includes a modulation/demodulation (MODEM) device (not shown) coupled to PSTN extension 104. However, a client computer

communication system illustrated in Figure 1, is significantly more complex than that which is depicted. For example, each SSP of PSTN 140 may service thousands of PSTN extensions, and there are numerous SSPs, STPs and SCPs in a common PSTN implementation. Internet 150 includes well over several hundred thousand networks. Together, PSTN 140 and internet 150 interconnect millions of client computers and web servers. Nonetheless, Figure 1 does capture a number of the more relevant components of a communication system necessary to illustrate the interrelationship between client computer 102, web server 128, bridgeports 162 and 168, and handset 142, such that one skilled in the art may practice the present invention. Also, while the present invention is being described in the context of client computer 102 being engaged in data communication with web server 128, as will be readily apparent from the description to follow, the present invention may be practiced with any "client" computer engaged in data communication with any "web" or "info" server.

Turning now to Figure 2A, a flow chart illustrating one embodiment of the method steps of the present invention for establishing and facilitating a voice call to a PSTN extension for a networked client computer is shown. For ease of explanation, the method of Figures 2A-2C will be developed in the context of an example implementation, wherein a user of client computer 102 is engaged in a data communication session involving a webpage, projected by web server 128, which incorporates a Push-To-Talk™ feature wherein the user of the webpage may "push" a displayed Push-To-Talk™ button to cause a voice connection to be established with a local office, retail center and the like, associated with the web server, enabling the user of client computer 102 to engage in a voice call with a user of a PSTN endpoint (e.g., handset 142) located at the "local office". [Those skilled in the art will appreciate that the terms "push" and "pushing" are metaphoric descriptions of the action taken by a user of the client computer. The action is in actuality accomplished, e.g., by the user clicking a mouse button, upon moving a cursor over the displayed Push-To-Talk™ button.]

With reference to Figure 2A, the method begins at step 202 with a user of client computer 102 "pushing" the Push-To-Talk™ button projected with the webpage. As described earlier, the Push-To-Talk™ button is pre-associated with a bridgeport, e.g., page bridgeport 162. In one embodiment, the pre-association is accomplished via HyperText Markup Language (HTML) elements embedded in the webpage, identifying the Uniform Resource Locator (URL) of bridgeport 162. The HTML elements further specify that a Push-To-Talk™ event notice including the URL of web server 128 is to be posted to page bridgeport 162. Thus, in response to the user's "pushing" of the Push-To-Talk™ button, a HyperText Transmission Protocol (HTTP) connection is temporarily established between client computer 102 and page bridgeport 162, and a message posting the Push-To-Talk™ event is sent to page bridgeport 162.

For the illustrated embodiment, in response to the Push-To-Talk™ event notification, page bridgeport 162 identifies itself to client computer 102, providing client computer 102 with its internet protocol (IP) address, step 204. The HTTP connection is closed upon sending the return data to client computer 102 by page bridgeport 162. In one embodiment, the identification and provision of page bridgeport's IP address also includes identification that the information is associated with a Push-To-Talk™ button projected by web server 128. More specifically, the URL of web server 128 is also returned to client computer 102. For the illustrated embodiment, the returned data also includes a command for starting up the client bridgeport driver on client computer 102.

Next, for the illustrated embodiment, client computer 102 identifies itself to page bridgeport 162, providing page bridgeport 162 with its IP address and attributes. For the illustrated embodiment, the identification is performed by the launched client bridgeport driver, by way of another HTTP connection. In one embodiment, the identification is also made in the form of a call request to talk to a person, such as an agent, associated with web server 128. In one embodiment, the attribute information includes a zip code for the area

client computer 102 is located. In another embodiment, the attribute information includes a PSTN extension associated with client computer 102. In one embodiment, the URL of web server 128 is also re-transmitted.

In response, page bridgeport 162 determines a destination PSTN extension for the requested call, step 208. In one embodiment, the determination is based on attributes of web server 162, e.g. the URL or the associated corporate name (if provided) of web server 162. In another embodiment, the determination is also based on attributes of client computer 102, e.g. the zip code or the telephone area code/prefix associated with client computer 102. In one embodiment, page bridgeport 162 is equipped with a PSTN extension database having the necessary attributes and PSTN extension information for performing the determination. In an alternate embodiment, page bridgeport 162 is not equipped with such a PSTN extension database, but equipped with services that access external on-line services (e.g., geographic location services, directory services, etc.) to make the determination. An example of an on-line geographic service is MapBlast™, developed by Vicinity Corporation of Palo Alto, CA.

Upon determining the destination PSTN extension, for the illustrated embodiment, page bridgeport 162 selects an internet/PSTN changeover server. In one embodiment, the internet/PSTN changeover server is a bridgeport, such as bridgeport 165 (hereafter, changeover bridgeport), where the requested voice call would be routed off internet 150 and onto PSTN 140, step 210. The selection is made from a "community" of bridgeports, to be described more fully below. In one embodiment, the "community" of bridgeports are "private" bridgeports deployed by the corporation of web server 128. In another embodiment, the "community" of bridgeports are "public" bridgeports deployed by a service company that offers "bridgeport" service of the present invention, and subscribed by the corporation of web server 128. Note that page bridgeport 162 may select itself as the changeover bridgeport, either because of the selection criteria employed dictate the result, or by virtue of a singleton community, i.e., page bridgeport 162 is the only bridgeport of the "community". For ease of

understanding, the remaining descriptions will be presented in the context of bridgeport 165 being the selected changeover bridgeport.

Upon selecting changeover bridgeport 165, for the illustrated embodiment, page bridgeport 162 registers the requested call with changeover bridgeport 165, step 210. In one embodiment, the registration reserves bandwidth on changeover bridgeport 165 for the requested call. In one embodiment, the registration includes provision of the source IP address of the voice call, i.e., the IP address of client computer 102, the source type (e.g. H.323), the destination address, i.e. the destination PSTN extension of handset 142, and the destination type (e.g. POTS).

Next, for the illustrated embodiment, page bridgeport 162 identifies changeover bridgeport 165 to client computer 102, providing client computer 102 with the IP address of hop off bridgeport 165, step 212. In one embodiment, where the call request is made through an HTTP connection, step 212 also includes closing the HTTP connection. In one embodiment, the identification also includes provision of the URL of web server 128. In one embodiment, both the identification, i.e., the IP address of changeover bridgeport 165, and the URL of web server 128 are provided to client bridgeport driver.

Next, for the illustrated embodiment, client computer 102 places a net call to changeover bridgeport 165, step 214. In one embodiment, the net call is an H.323 call placed by an internet telephony application. In one embodiment, step 214 also includes automatic launching of the internet telephony application to place the net call, if an internet telephony application has not been previously launched. For the illustrated embodiment, the automatic launching is performed by the client bridgeport driver.

In response, changeover bridgeport 165 places a voice call to the PSTN extension of handset 142, and bridges the net and the voice calls, step 216, thereby allowing a user of

client computer 102 to communicate with a user of handset 142. In bridging the two calls, changeover bridgeport 165 digitizes and compresses inbound call signals received from handset 142, and delivers the encoded call signals to client computer 102 via the previously established H.323 connection. The compressed inbound call signals are decompressed by the communication interface of client computer 102 and rendered by the internet telephony application. Similarly, outbound call signals emanating from client computer 102 are digitized by the audio interface, compressed by the communication interface of client computer 102 and delivered to changeover bridgeport 165 via the H.323 connection, wherein they are decompressed, and upon conversion, forwarded to handset 142. In other words, changeover bridgeport 165 converts the voice information between PSTN and IP protocols and delivers voice call information to/from handset 142 and client computer 102 until call completion.

With reference now being made to Figure 2B, one embodiment of step 216 is illustrated. As the H.323 connection between client computer 102 and changeover bridgeport 165 is established, changeover bridgeport 165 determines if a direct connection with client computer 102 is to be established, if client computer 102 and changeover bridgeport 165 are not in direct connection with each other. If changeover bridgeport 165 makes the determination not to establish a direct connection between client computer 102 and itself, changeover bridgeport 165 establishes and facilitates the voice call to PSTN extension 143 as described earlier, in step 230. Additionally, in those cases where changeover bridgeport 165 is not in direct telephony contact with client computer 102, (i.e., the call is being routed through a number of intermediate servers throughout Internet 150), changeover bridgeport 165 also monitors the "quality" of the voice call. Any number of metrics known in the art may be beneficially employed to measure the quality of the voice call, e.g. the number of audio packets dropped within a predetermined time interval (Packet Error Rate), the signal to noise ratio, the rise in the measured noise floor, and the like.

If quality of the voice call is not monitored and the voice call has not been completed, steps 232 - 234, changeover bridgeport 165 returns to step 230. However, if quality of the voice call is monitored, step 232, changeover bridgeport 165 further determines if the quality has reached a predetermined threshold and, in addition, whether the user of client computer 102 has previously rejected an offer to switch to a direct telephony connection with changeover bridgeport 165, step 236. If the measured quality level is in an acceptable range, or the user has previously rejected an offer to establish a direct connection, changeover bridgeport 165 returns to step 230. Otherwise, changeover bridgeport 165 prompts the user as to whether the user would like to switch to a direct telephony connection between client computer 102 and changeover bridgeport 165, step 238. If the user responds with an election not to switch to a direct telephony connection, step 240, changeover bridgeport 165 returns to step 230. However, if the user responds with an election to switch to a direct telephony connection, step 240, changeover bridgeport 165 causes a direct telephony connection to be established between itself and client computer 102, step 242. Upon establishing the direct telephony connection, changeover bridgeport 165 continues to facilitate the voice call as described earlier, step 230.

Similarly, if back in step 228, changeover bridgeport 165 makes the determination to establish a direct connection between client computer 102 and itself, the process proceeds to step 238 wherein the user of client computer 102 is offered the opportunity to accept or decline to establish a direct connection between client computer 102 and changeover bridgeport 165. From step 238, the process proceeds as described earlier. Whether changeover bridgeport 165 proceeds to step 230 or to step 238 directly from step 228 will be described in more detail below.

Turning now to Figure 2C, a flow chart illustrating one embodiment of a method for establishing a direct telephony connection between client computer 102 and the PSTN extension associated with handset 142. In the context of the illustrated embodiment,

changeover bridgeport 165 provides a software to client computer 102, step 250. The software is a set of instructions containing information necessary to enable client computer 102 to establish a direct telephony connection with changeover bridgeport 165. Upon receipt of the software by client computer 102, the software tears down the existing connection between client computer 102 and the ISP currently in direct telephony contact with client computer 102 (e.g., ISP 112), step 252. Having disconnected the prior connection, the provided software initiates a direct telephony connection to changeover bridgeport 165 through the modem of client computer 102, thereby eliminating any intermediate servers, step 254. For the illustrated embodiment, the software is "equipped" with one of changeover bridgeport's PSTN extensions before being provided to client computer 102. Finally, once the direct telephony connection from client computer 102 is established, changeover bridgeport 165 logically "re-links" client computer 102 with the PSTN extension of handset 142, step 256, thereby allowing the previously established voice call to be re-connected (although, it should be noted that the physical PSTN connection between the changeover bridgeport and the PSTN extension is never disconnected; rather, it is the data connection between the client computer and the changeover bridgeport that is disconnected in favor of a direct telephony connection). In an alternate embodiment, the direct connect software may be pre-installed on client computer 102, e.g. as part of client bridgeport application described earlier.

Returning now to step 228 of Figure 2B, in one embodiment, changeover bridgeport 165 proceeds to step 238 directly, without first monitoring the quality of the voice call, if the changeover bridgeport 165 "knows" *a priori* that the quality of the voice call will be below an acceptable threshold. For example, assume that it has been determined that the quality of a voice call drops below a predetermined quality level when the number of intermediate servers between a client computer (e.g., client computer 102) and a changeover bridgeport (e.g., changeover bridgeport 165) exceeds a certain number N_2 . Given this example, notwithstanding the fact that changeover bridgeport 165 was originally selected as the "best"

changeover bridgeport from the "community" of bridgeports, if the actual number of intermediate servers N_1 between client computer 102 and changeover bridgeport 165 exceeds N_2 , changeover bridgeport 165 will know, before the voice call has been established, that the quality of the voice call will be below the acceptable predetermined quality level. In one embodiment, the number of intermediate servers is returned to changeover bridgeport 165 when page bridgeport 162 selects bridgeport 165 as the changeover bridgeport and registers the voice call with changeover bridgeport 165. In one embodiment, changeover bridgeport 165 preserves the inputs along with the registration information for use at step 228. In another embodiment, bridgeport 165 may retain up to a predetermined number of "bid responses", which will include the number of intermediate servers, for a period of time sufficiently long to ensure that if bridgeport 165 is selected as the changeover bridgeport, the bid response information is available.

In an alternate embodiment, changeover bridgeport 165 proceeds to step 238 directly, without having first monitored the quality of the voice call, if changeover bridgeport 165 was instructed to do so by page bridgeport 162 when page bridgeport 162 selects bridgeport 165 as the changeover bridgeport and registers the voice call. Page bridgeport 162 may select to so instruct changeover bridgeport 165, based on a similar decision process described earlier for the case where changeover bridgeport 165 makes the decision. Alternatively, page bridgeport 162 may do so because it is so instructed by web server 128, through parameters embedded within the Push-To-Talk™ button. In other words, the Push-To-Talk™ button, projected by web server 128, may be a "direct quality" Push-To-Talk™ button that inherently includes offering the user of client computer 102 with the option to conduct the voice call via a direct connection at a "direct quality" level.

In summary, page bridgeport 162, in response to a Push-To-Talk™ event notification, solicits inputs and selects a changeover bridgeport (e.g., bridgeport 165) from a "community" of bridgeports. Changeover bridgeport 165, in response to the placement of a net call from

client computer 102, establishes a PSTN connection to the appropriate PSTN extension and bridges the call. In so doing, client computer 102 may be offered to have the voice call conducted at a "direct quality" level via a direct connection between client computer 102 and changeover bridgeport 165. The offer may be presented dynamically in the middle of the call, at the discretion of changeover bridgeport 165 as it detects the quality of the voice call dropping below an predetermined quality level. Alternatively, the offer may be presented at the beginning of the voice call, at the discretion of changeover bridgeport 165, page bridgeport 162 or web server 128.

Note that except for the solicitation of the concurrence of the user of client computer 102 to switch to a direct telephony connection with changeover bridgeport 165, steps 204 - 206 are all performed automatically in response to step 202, without requiring any intervention from the user of client computer 102. In particular, it does not require the user of client computer 102 to enter the PSTN extension of handset 142, nor the IP address of changeover bridgeport 165. It does not even require the user of client computer 102 to know this information. All that is required of the user is the metaphorical "pushing" of the Push-To-Talk™ button projected by web server 128. Furthermore, it should also be noted that the concurrence of the user of client computer 102 is not necessary. For example, in the instance where the direct connection will be toll free, e.g., a local call, or the changeover bridgeport has an "800" access number, or the toll charge is to be borne by the web server, the Push-To-Talk™ button may be embedded with direct connection software that automatically establishes a direct connection with the changeover bridgeport.

It should also be noted that the voice connection has minimal impact on establishing any additional data connections with any number of web servers 120 and 128. In other words, client computer 102 may continue to browse webpages offered by web servers 120 and 128, while simultaneously supporting the voice connection with handset 142 via changeover bridgeport 165. In addition, although there may be a number of intermediate

routers in Internet 150 between changeover bridgeport 165 and client computer 102, changeover bridgeport 165 is the only server charged with supporting both the voice connection and the H.323 connection, while the voice connection is transparent to the intermediate routers. In other words, the information exchange between changeover bridgeport 165, through the plurality of intermediate routers of Internet 150, to client computer 102 will appear as normal data packets to the intermediate routers.

Returning now to the selection of a changeover bridgeport described in step 210, page bridgeport 162 selects changeover bridgeport 165 from a community of bridgeports. In one embodiment, page bridgeport 162 first solicits input on a number of call characteristics from each bridgeport member of the community. The call characteristics may include the number of intermediate servers (Is) required to connect client computer 102 with the responding bridgeport member, the toll charge (Tc) that may be incurred by placing the telephone call from the responding bridgeport, the bandwidth (B) currently available on the responding bridgeport, the number of PSTN connections (P) supported by the responding bridgeport, service premiums (S), if any, charged by the responding bridgeport, and so forth. In the context of the example implementation, one member bridgeport may respond with an indication that there are no intermediaries between itself and client computer 102 which may provide a higher quality of service, however, given its connection point to PSTN 140, there may be a significant toll charge incurred in placing the telephone call to handset 142 from this member bridgeport. On the other hand, another member bridgeport may respond with a low toll charge, but with a higher number of intermediaries as the voice data will have to route through a large number of routers.

In any event, for the illustrated embodiment, page bridgeport 162 calculates a Call Metric (CM) for each of the responding bridgeport that is representative of the bridgeports ability to establish and facilitate the voice call between client computer 102 and handset 142. Equation (1) is an example of an equation used to calculate the Call Metric for each of the

responding bridgeport, wherein the bridgeport with the lowest CM is determined to be able to provide the best all around service. It should be noted that equation (1) is merely illustrative, as one skilled in the art will appreciate that suitable alternative equations may be beneficially employed to calculate alternative call metrics.

$$CM_i = W_1(Is_i) + W_2(Tc_i) + \frac{W_3}{B_i} + \frac{W_4}{P_i} + W_5(S_i) \quad (1)$$

where: - Is number of intermediate servers
 Tc toll charge estimate
 B available bandwidth on responding server
 P number of PSTN ports available on responding server
 S premium service charge
 W weighting factor

In one embodiment of the present invention, the number of intermediate servers (Is) is determined by each of the responding bridgeports from the community of bridgeports through the use of a "traceroute" function, common to the UNIX network operating environment. As one skilled in the art will appreciate, the execution of a "traceroute" command by a bridgeport will produce a result quantifying the number of intermediate routers between the execution bridgeport and a destination address (provided in the command line). Accordingly, in one embodiment, the execution of the command: "traceroute (IP_address)", will return a number representative of the intermediate routers required for the responding bridgeport to communicate with the source address, represented by IP_address.

In one embodiment, an estimate of the toll charge is determined locally at the responding bridgeport by accessing a toll rate table stored on the responding bridgeport. An example toll rate table is depicted in Table 1. As illustrated in the example toll rate table of Table 1, an estimate toll charge is determined by analyzing the components of the destination telephone number against a toll charge hierarchy. If none of the elements of the destination

telephone number match (i.e., hit) the elements of the toll rate table, a high toll charge estimate is returned. If, however, there is a hit on the area code and the telephone number prefix, a much lower toll charge or even toll free estimate is returned. In alternate embodiments of the present invention, the responding bridgeport may query the local SSP (i.e., the SSP to which it is coupled) to ascertain a specific toll charge value. In such a case, the communication between the responding bridgeport and the local SSP is conducted via an out-of-band signaling protocol such as Signaling System 7 (SS7).

<u>Elements of Destination Telephone No.</u>	<u>Relative Toll Charge Estimate</u>
No Hit	\$ \$ \$ \$
Area Code Hit	\$ \$
Area Code and PSTN Prefix Hit	0

TABLE 1: TOLL CHARGE LOOKUP TABLE

In an alternate embodiment, page bridgeport 162 may also involve client computer 102 in the selection of a changeover bridgeport, by presenting the solicited responses to the client computer 102, in either an uncondensed or a condensed format. Whether client computer 102 should be involved in the selection process is a design choice, a trade off between ease of use (without requiring intervention from the user of client computer 102) and functionality (allowing the user of client computer 102 to veto or influence the selection). Between the two distinct choices, a number of alternate hybrid embodiments exist which do not deviate from the scope and spirit of the present invention.

Although bridgeport 162 has been described as a page bridgeport, and bridgeport 165 has been described as a changeover bridgeport, it should be noted that since they are both members of a community of bridgeports they may, at one time be a page bridgeport, while

serving as a changeover bridgeport at another time, or the bridgeport may simultaneously server as both the page and changeover bridgeport. As will be readily apparent from the descriptions to follow, bridgeports 162 and 165 of the present invention can be practiced with computer servers programmed to perform the above described bridgeport functions, thus it is expected that a bridgeport may be integrated with other equipment in a variety of manners, for examples, with a web server, an ISP, a STP, and so forth.

Turning now to **Figures 3 and 4**, two block diagrams illustrating the hardware and software elements of an exemplary computer server 300 suitable to be employed as a bridgeport are depicted. As illustrated, exemplary computer server 300 is comprised of multiple processors 302a - 302n and memory subsystem 308 coupled to processor bus 304 as depicted. Additionally, computer server 300 is comprised of a second bus 310, a third bus 312 and a fourth bus 314. In one embodiment, buses 312 and 314 are Peripheral Component Interconnect (PCI) buses, while bus 310 is an Industry Standard Architecture (ISA) bus. PCI buses 312 and 314 are bridged by bus bridge 316, and bridged to ISA bus 310 and processor bus 304 by I/O controller 306. Coupled to PCI bus 312 are network interface 318 and display interface 320, which in turn is coupled to display 322. Coupled to PCI bus 314 is computer telephony interface (CTI) 324, PSTN interface 326 and SS7 Interface 328. Coupled to ISA bus 310 are hard disk interface 330, which in turn is coupled to a hard drive 332. Additionally, coupled to ISA bus 310, keyboard and cursor control device 334, which in turn is coupled keyboard 336 and cursor control device 338.

CTI interface 324 provides the necessary hardware to interface exemplary computer server 300 to telephony equipment, such as private branch exchange (PBX) equipment. PSTN interface 326 provides the necessary hardware to interface exemplary computer server 300 to a plurality of PSTN communication lines (e.g., T1, E1 or POTS), wherein the actual number of PSTN communication lines interfaced will be implementation dependent. Additionally, PSTN interface 326 provides advanced DSP-based voice, dual-tone multiple

frequency (DTMF) and call progress functionality, which allows for downloadable DSP protocol and voice processing algorithms, thereby providing CODEC support locally on the interface. Examples of supported codecs include the Global System for Mobile Communications (GSM) codec and the ITU-T G.723.1 protocol codecs, the specification for which are commonly available from the GSM consortium and the International Telecommunications Union, respectively. Similarly, SS7 interface 328 provides the hardware necessary to interface exemplary computer server 300 with PSTN trunk lines (e.g., ISDN) supporting the out-of-band communication protocol (e.g., SS7)) used between PSTN network elements (i.e., SSP-SSP, SSP-STP, STP-SCP, etc.). In one embodiment, PSTN interface 326 is preferably an AG-T1™ (for U.S. implementations, while an AG-E1 may be seamlessly substituted for European implementations), while SS7 interface 328 is preferably the TX3000™, both of which, along with their accompanying software drivers, are manufactured by and commonly available from Natural Microsystems of Natick, Massachusetts. Otherwise, all other elements, processors 302*, memory system 308 and so forth perform their conventional functions known in the art. Insofar as their constitutions are generally well known to those skilled in the art, they need not be further described.

From a software perspective, Figure 4 illustrates the software elements of exemplary computer server 300. In particular, exemplary computer server 300 is shown comprising an application layer consisting of a Bridgeport Management Driver 402, Hop Off™ driver 404, and other drivers 406. Hop Off™ driver 404, supported by Management Driver 402, optional drivers 406, and abstracted service layer 408 implements the method steps of Figure 2 that are the responsibility of the community of bridgeports (i.e., bridgeports 162, and 165). In this respect, a changeover bridgeport appropriately configured may be referred to as a Hop Off™ bridgeport, in view of its incorporation of Hop Off™ driver 404.

* Hop Off™ is a Trademark of eFusion™, Incorporated of Beaverton, Oregon.

The Service Abstraction Layer (SAL) 408 is shown comprising SS7 services 410, CTI Services 411, Management Services 412, Connection Services 414, Streaming Services 416, and Data Services 418. The protocol/service layer is shown comprising Telephony Application Programming Interface (TAPI) 420, Telephony Connection Protocol 422, PSTN Data Interface 424, CODEC 426, Real Time (Streaming) Protocol 428, and HTTP server 434. Also shown in this "layer" are configuration management data 419 maintained by management service 412, and codec services 426 employed by streaming services 416. The driver layer is shown comprising SS7 driver 427, CTI driver 429, PSTN driver 430 and socket service 432. Data and control information are exchanged between these elements in the fashion depicted.

Within the context of the present invention, one purpose of SAL 408 is to provide an Application Programming Interface (API) for all the available bridgeport and related services in exemplary computer server 300. The API abstracts out the actual modules used for providing services such as connection establishment (414), streaming and data exchange services (416 and 418). Additionally, SAL 408 provides the common operation tools such as queue management, statistics management, state management and the necessary interface between the software services (i.e., drivers in the driver layer). SAL 408 is also responsible for loading and unloading the appropriate drivers as appropriate.

Connection service 414 includes a connection establishment and tear-down mechanism facilitating the interconnection to the PSTN 140. Additionally, for the illustrated embodiment, connection service 414 employs connection and compatibility services which facilitate interoperation between communication equipment that support industry standards, thereby allowing a variety of communication equipment manufactured by different vendors to be benefited from the present invention. Connection services 414 include, in particular, services for supporting standard video telephony, such as ITU-T's H.323 video telephony, and standard data communication, such as ITU-T's T.120 data communication protocol.

Examples of the connection establishment and tear-down mechanisms supported by connection service layer 414 include opening and starting PSTN ports, call control, DTMF collection, and tone generation, to name but a few.

Streaming service 416 is responsible for interfacing with the components that provide the real-time streaming functionality for the multimedia data. Once the connection has been established between the connection points (i.e., PSTN, H.323, etc.), streaming service 416 will take over the management and streaming of data between the two connected parties, until the connection is terminated. CODEC service 426 facilitates the above described compression and transmission of inbound call signals from handset 142 as well as decompression and transmission of outbound call signals from client computer 102.

Data service 418 is responsible for providing non real-time peer to peer (i.e., computer-computer) messaging and data exchange between exemplary computer server 300 and other Internet and perhaps PSTN based applications. Sending messages to exemplary computer server end-points (i.e., other similarly equipped bridgeports on the Internet) or other servers within the PSTN is accomplished via data service 418.

CTI services 411 service all communications and automatic call distribution (ACD) necessary for Private Branch Exchange (PBX) based systems. SS7 services 410 service all out of band communications with STPs and/or SCPs of PSTN 140.

PSTN driver 430 is equipped to accommodate particularized PSTN interfaces 326, whereas CTI driver 429 is equipped to support particularized ACD and PBX equipment. Similarly, SS7 driver 427 is equipped to support particularized SS7 interface 328.

Turning now to Figures 5, 6 and 7, graphical representations of alternate embodiments of a Push-To-Talk™ indicator for requesting a voice communication session,

suitable for use in the exemplary communication system of Figures 1-4, is shown. In Figure 5, indicator 500 is a binary prompt embedded in the HTML information for a webpage projected by a web server (e.g., web server 128), and is selectively actuated by a user to initiate a voice connection with a PSTN extension associated with a handset. In one embodiment, indicator 500 is a Push-To-Talk™ button. In response to a user actuation of Push-To-Talk™ button 500, a prompt 502 for location information is projected to the user of client computer 102. In one embodiment, prompt 502 may be a pull-down selection of predetermined locations (e.g., city/state and country locations), wherein the user is requested to make a selection of a location closest to the user. In response to providing the location information, an additional prompt 504 may then be generated which allows the user to set the weighting of different characteristics associated with establishing and facilitating the voice connection (e.g., cost and call quality). In one embodiment, call characteristic prompt 504 includes slide members 506 and 508 which are selectively manipulated by the user of client computer 102 to correspond to "weighting values" (W_i) to be given to, for example, cost and connection quality, respectively.

Alternatively, in Figure 6, indicator 600 is shown comprising Push-To-Talk™ button 500 and location information prompt 502. In response to the coextensive selection of location information and the actuation of Push-To-Talk™ button 500, an additional prompt 504 may then be generated which allows the user to set the weighting of different characteristics associated with establishing and facilitating the voice connection (e.g., cost and call quality). In the alternate embodiment of Figure 7, exemplary indicator 700 is shown comprising Push-To-Talk™ button 500, location information prompt 502 and call characteristic prompt 504. In accordance with the coextensive selection of location information and call characteristic information with the actuation of Push-To-Talk™ button 500, the establishment of a voice connection is initiated in accordance with the method steps of Figures 2A-2c.

While the method and apparatus of the present invention has been described in terms of the above illustrated embodiments, those skilled in the art will recognize that the invention is not limited to the embodiments so described. In particular, the present invention may beneficially be implemented in combination with other technology to accommodate a wide variety of communication needs. For example, in one embodiment, the Push-To-Talk™ service instead of being offered by the corporate web server, may be offered by an independent third party. Accordingly, an intermediate third party server (or bridgeport) other than the web server may "inject" a Push-To-Talk™ button into the information exchange between the web server and the client computer which, when enabled, will initiate a voice call to the local affiliate. Just such an invention is discussed in copending U.S. Patent Application No. <insert patent application number>, docket number 002784.P001 entitled "Method and Apparatus for Value Added Content Delivery", filed contemporaneously herewith and commonly assigned to the assignee of the present application, the disclosure of which is fully incorporated herein by reference. In other communication environments, it may be beneficial for the user of the client computer and the user of the PSTN extension (who also has access to a computer) to view common information on the computer displays while maintaining the voice connection (e.g., a customer service, help desk, sales, etc.). A description of just such an invention is discussed in the U.S. patent application entitled "Method and Apparatus for Synchronizing Information Browsing Among Multiple Systems", U.S. Patent Application No. <insert patent application number>, attorney docket number 002784.P004, filed contemporaneously herewith and commonly assigned to the assignee of the present application, the disclosure of which is fully incorporated herein by reference. In other embodiments, it may be beneficial to provide customized call processing services to the communication environment. Examples of such customized call processing services include providing an automated call attendant service having specialized call processing features. Just such an invention is disclosed in the U.S. patent application entitled "Method and Apparatus for Providing Customized Call Processing", U.S. Patent Application No. <insert patent application number>, attorney docket number 002784.P010, filed contemporaneously

herewith and commonly assigned to the assignee of the present application, the disclosure of which is fully incorporated herein by reference. All of the foregoing patent applications are expressly incorporated herein by reference. Thus, the present invention can be practiced with modification and alteration within the spirit and scope of the appended claims. Accordingly, the description is to be regarded as illustrative instead of restrictive on the present invention.

Thus, a method and apparatus for establishing a voice call to a PSTN extension for a networked client computer, and for routing the call off of the network has been described.

CLAIMS

We claim:

1. An apparatus comprising:
a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for establishing and facilitating a voice call to a PSTN extension for a client computer, wherein the communication services include services for receiving an event notification from the client computer denoting the client computer's acceptance of an offer by a server to establish a voice communication session, and determining the PSTN extension in accordance with one or more attributes of the server; and
an execution unit coupled to the storage medium for executing the plurality of programming instructions.
2. The apparatus of claim 1, wherein
the event notification includes identification information of the server, and
the communication services determine the PSTN extension in accordance with the identification information.
3. The apparatus of claim 2, wherein the communication services further include services for accessing a geographic location service, and the communication services access the geographic location service to determine a suitable voice call destination.
4. The apparatus of claim 2, wherein
the server is a web server having a URL,
the event notification includes the URL of the web server, and
the communication services determine the PSTN extension in accordance with the URL of the web server.

5. The apparatus of claim 4, wherein the apparatus further includes a database of PSTN extensions associated with a plurality of URLs of web servers, and the communication services access the database to look up the PSTN extension.
6. The apparatus of claim 1, wherein
the communication services further include services for providing the client computer with a network address of the apparatus, and receiving a supplemental voice call request from the client computer, including a network address and one or more attributes of the client computer, and
the communication services perform said determination of the PSTN extension further in accordance with one or more of the included attributes of the client computer.
7. The apparatus of claim 6, wherein the communication services further include services for causing a packet/voice bridging driver to be launched on the client computer.
8. The apparatus of claim 6, wherein
the one or more attributes of the client computer include a zip code of an area where the client computer is located, and
the communication services perform said determination of the PSTN extension taking into consideration the zip code.
9. The apparatus of claim 6, wherein
the one or more attributes of the client computer include a telephone number associated with the client computer, and
the communication services perform said determination of the PSTN extension taking into consideration at least the area code of the telephone number.

10. The apparatus of claim 1, wherein the communication services further include services for soliciting inputs from one or more other apparatuses on one or more operating characteristics associated with establishing the voice call to the PSTN extension, selecting either itself or one of the other apparatuses to place the voice call to the PSTN extension, and identifying the selected apparatus to the client computer.
11. The apparatus of claim 10, wherein the communication services further include services for registering the voice call to the PSTN extension with the selected apparatus.
12. The apparatus of claim 11, wherein the communication services further include services for receiving registration of the voice call to the PSTN extension.
13. The apparatus of claim 10, wherein the communication services further includes services for causing a packet based telephony application to be launched on the client computer.
14. The apparatus of claim 1, wherein the communication services further include services for receiving a packet based phone call from the client computer, placing the voice call to the PSTN extension, and bridging the packet based and voice calls.
15. The apparatus of claim 14, wherein the bridging services include services for converting and delivering call signals between the client computer and the PSTN extension.
16. The apparatus of claim 1, wherein the communication services further include telephony services for cooperating with telephony equipment coupled to the apparatus.
17. The apparatus of claim 1, wherein the storage medium further having stored therein programming instructions for implementing web server functions.

18. The apparatus of claim 1, wherein the storage medium further having stored therein programming instructions for implementing internet service provider functions.
19. A method for placing a voice call to a PSTN extension comprising the steps of:
- (a) receiving an event notification from a client computer denoting the client computer's acceptance of an offer by a server to establish a voice communication session, and
 - (b) determining the PSTN extension in accordance with one or more attributes of the server.
20. The method of claim 19, wherein
- the step of receiving an event notification includes receiving identification information of the server, and
 - the step of determining the PSTN extension is performed in accordance with the received identification information of the server.
21. The method of claim 20, wherein the step of determining the PSTN extension includes accessing a directory service to look up the PSTN extension using the received identification information.
22. The method of claim 20, wherein
- the server is a web server having a URL,
 - the step of receiving an event notification includes receiving the URL of the web server, and
 - the step of determine the PSTN extension is performed in accordance with the URL of the web server.
23. The method of claim 22, wherein the step of determining the PSTN extension includes accessing a database to look up the PSTN extension.

24. The method of claim 19, wherein

the method further comprises the steps of providing the client computer with a network address of the apparatus, and receiving a supplemental voice call request from the client computer, including a network address and one or more attributes of the client computer, and

the step of determining the PSTN extension is performed further in accordance with one or more of the included attributes of the client computer.

25. The method of claim 24, wherein the method further includes the step of causing a packet/voice bridging driver to be launched on the client computer.

26. The method of claim 24, wherein

the one or more attributes of the client computer include a zip code of an area where the client computer is located, and

the step of determining the PSTN extension includes taking into consideration the zip code.

27. The method of claim 24, wherein

the one or more attributes of the client computer include a telephone number associated with the client computer, and

the step of determining the PSTN extension includes taking into consideration at least the area code of the telephone number.

28. The method of claim 19, wherein the method further includes the steps of soliciting inputs from one or more apparatuses on one or more operating characteristics associated with establishing the voice call to the PSTN extension,

selecting one of the apparatuses to place the voice call to the PSTN extension, and identifying the selected apparatus to the client computer.

29. The method of claim 28, wherein the method further comprises the step of registering the voice call to the PSTN extension with the selected apparatus.
30. The method of claim 29, wherein the method further includes the step of receiving registration of the voice call to the PSTN extension.
31. The method of claim 28, wherein the method further includes the step of causing a packet based telephony application to be launched on the client computer.
32. The method of claim 19, wherein the method further includes the steps of receiving a packet based phone call from the client computer, placing the voice call to the PSTN extension, and bridging the packet based and voice calls.
33. The method of claim 32, wherein the step of bridging includes converting and delivering call signals between the client computer and the PSTN extension.
34. An apparatus comprising:
- a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for cooperating with a first bridging apparatus to establish and conduct a voice call with a PSTN extension, wherein the communication services include services for receiving an IP address of a second bridging apparatus, and in response, transmitting a call request for the voice call, including an IP address and other attributes of the apparatus for inclusion by the second bridging apparatus in determining the PSTN extension; and
 - an execution unit coupled to the storage medium for executing the plurality of programming instructions.

35. The apparatus of claim 34, wherein the attributes of the apparatus include a zip code of an area where the apparatus is located.
36. The apparatus of claim 34, wherein the attributes of the apparatus include a PSTN extension associated with the apparatus.
37. The apparatus of claim 34, wherein the communication services further include services for receiving an IP address of the first bridging apparatus, and launching a packet based telephony application to place a packet based phone call to the first bridging apparatus.
38. The apparatus of claim 37, wherein the first and the second bridging apparatuses are the same apparatus.
39. A method for cooperating with a first bridging apparatus to establish and conduct a voice call with a PSTN extension, the method comprising the steps of:
receiving an IP address of a second bridging apparatus, and
in response, transmitting a call request for the voice call, including an IP address and one or more attributes of a host apparatus implementing the method, for inclusion by the second bridging apparatus in determining the PSTN extension.
40. The method of claim 39, wherein the attributes include a zip code of an area where the host apparatus is located.
41. The method of claim 39, wherein the attributes include a PSTN extension associated with the host apparatus.
42. The method of claim 39, wherein the method further includes the steps of:
receiving an IP address of the first bridging apparatus, and

launching a packet based telephony application to place a packet based phone call to the first bridging apparatus.

43. The method of claim 42, wherein the first and the second bridging apparatuses are the same apparatus.

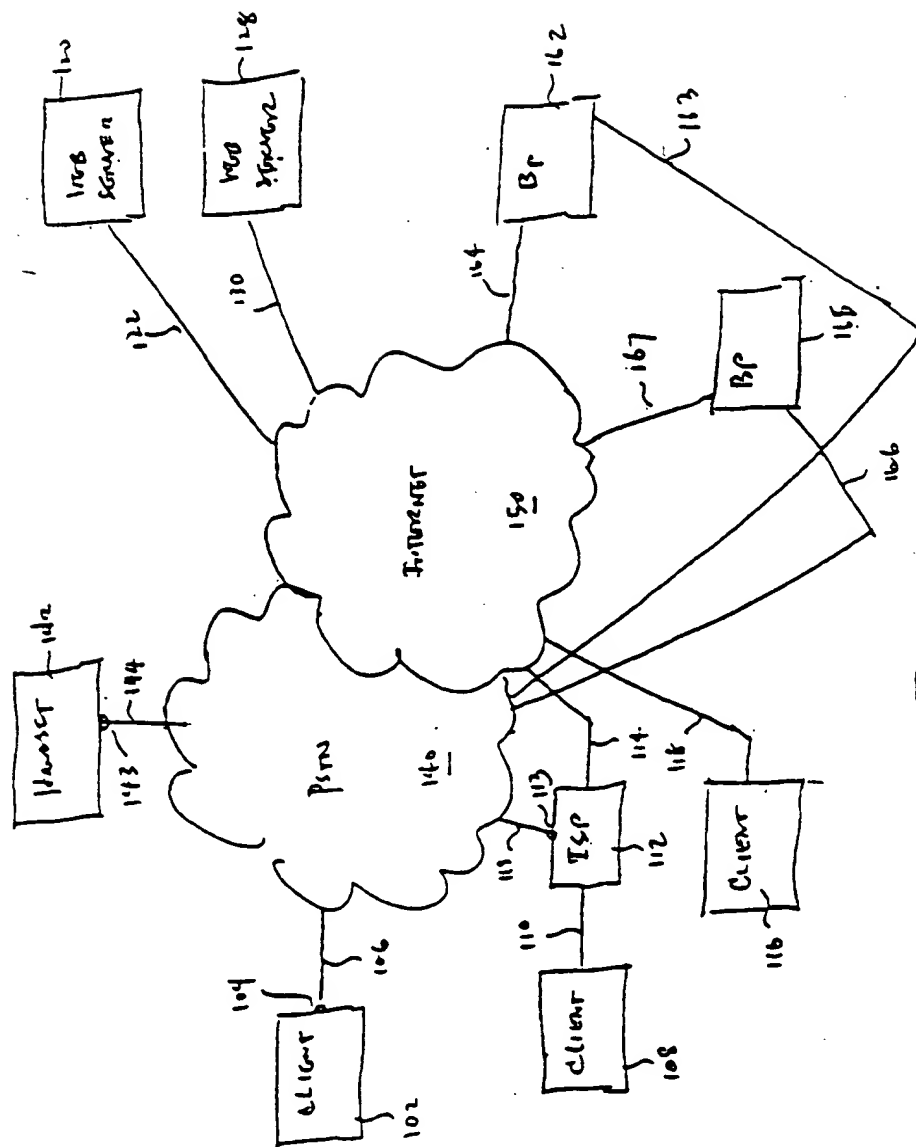
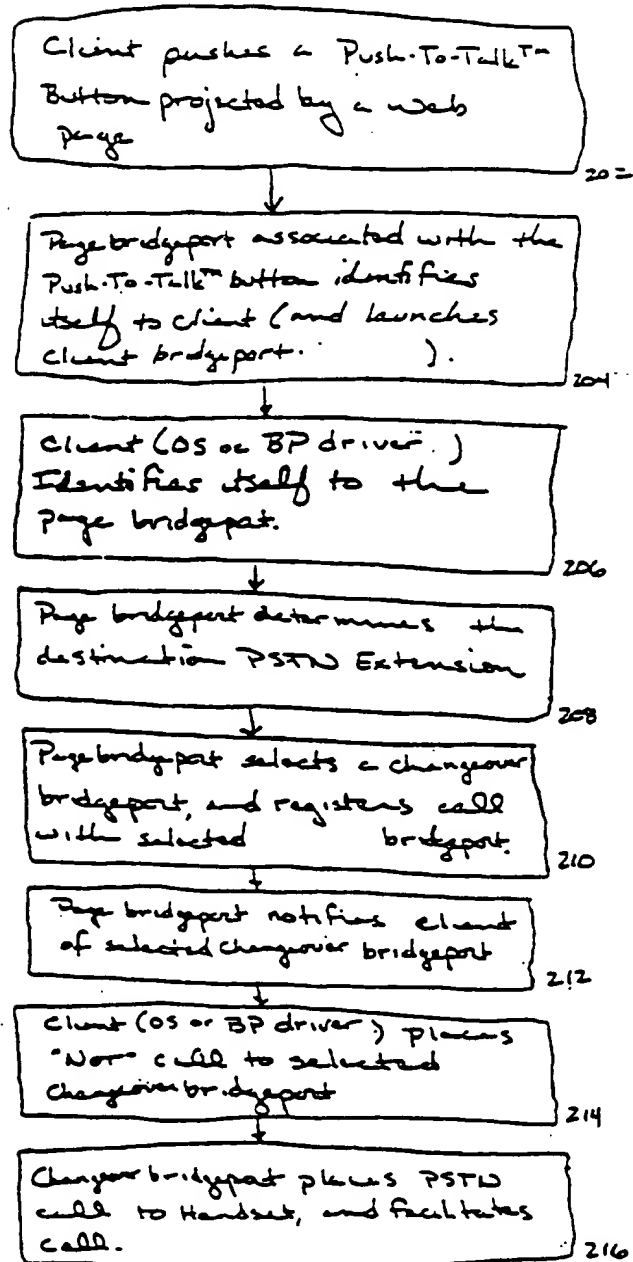


Figure 1

2/8

FIGURE 2A

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FIGURE 2B

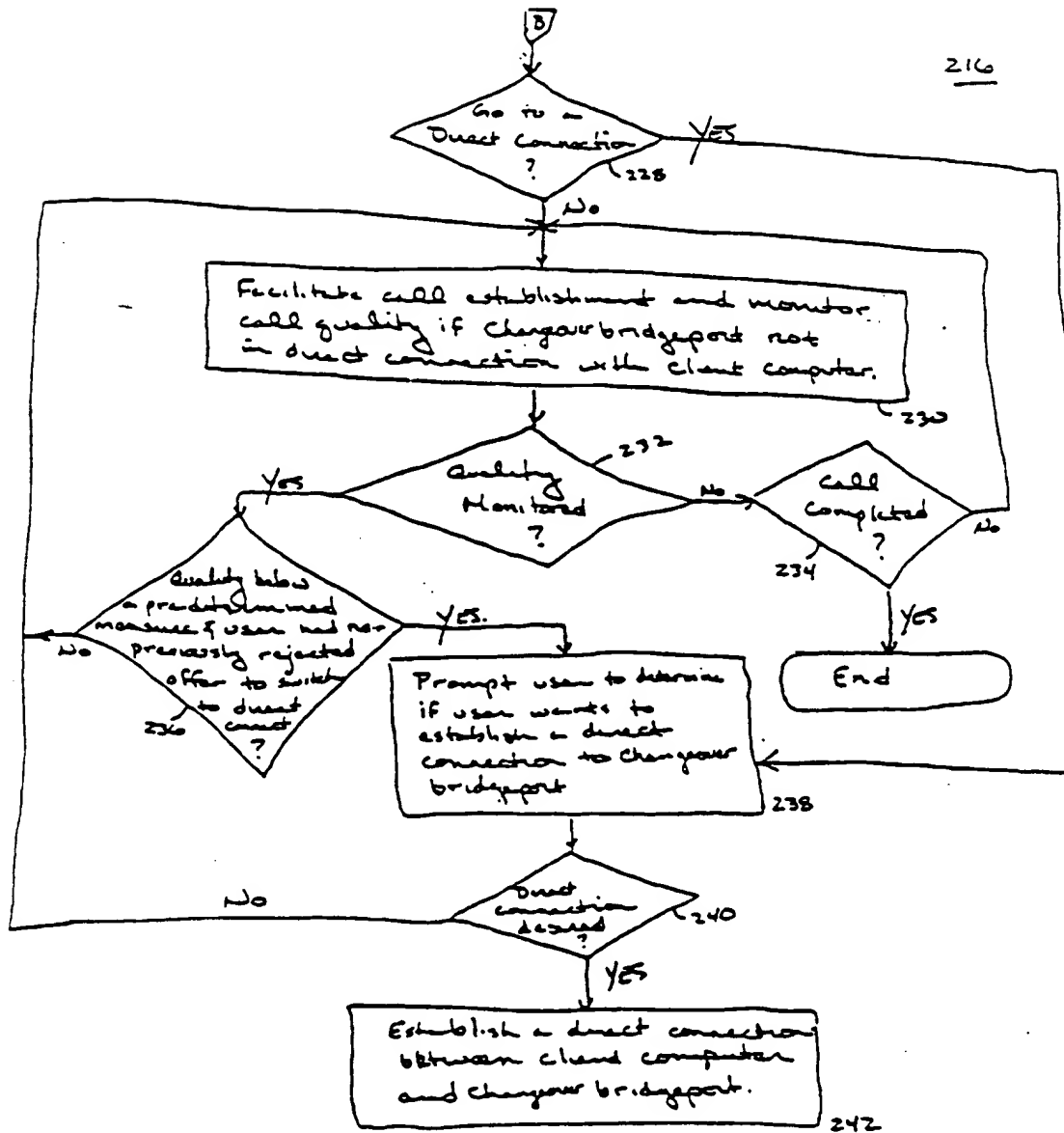


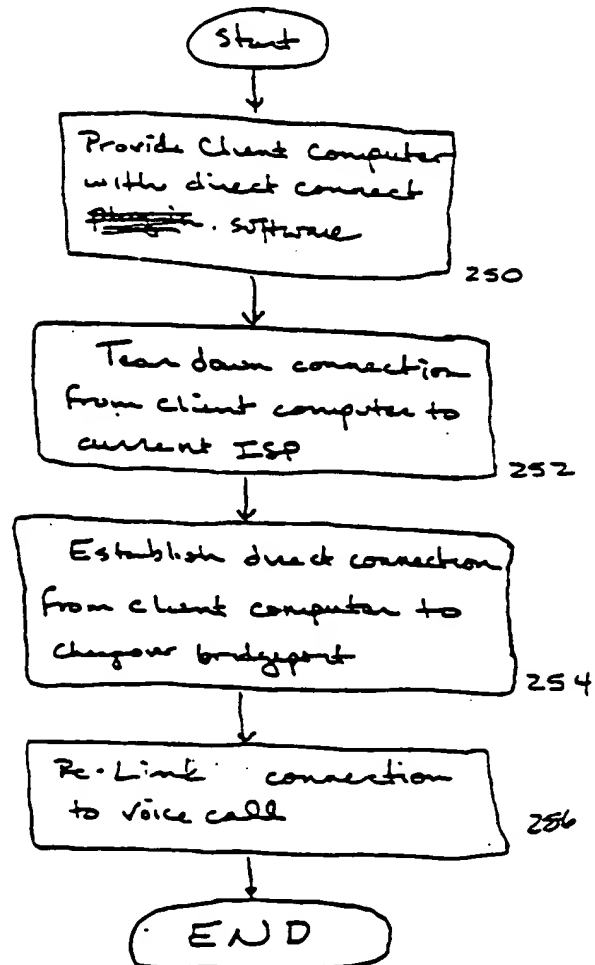
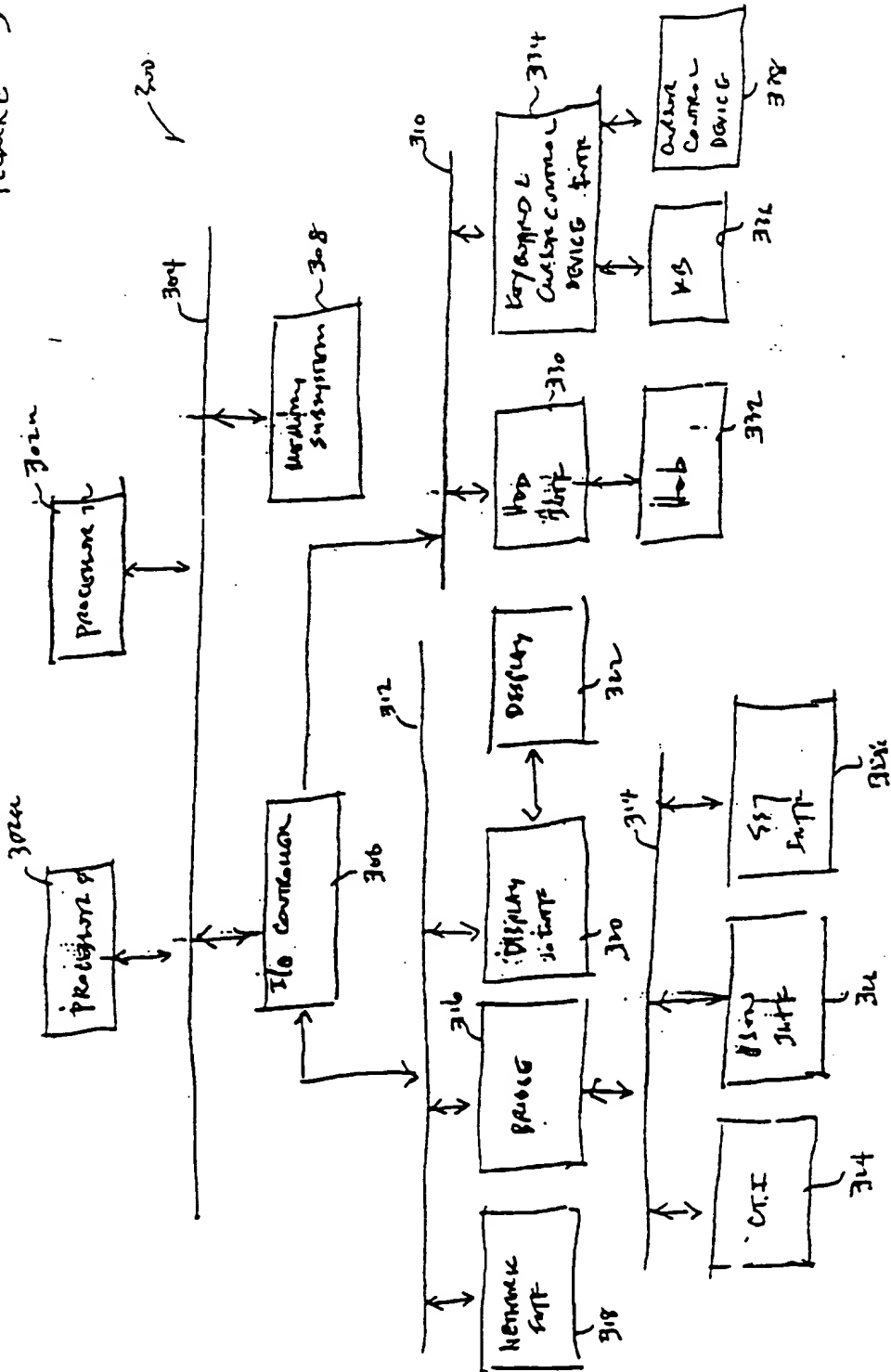
FIGURE 2C242

FIGURE 3



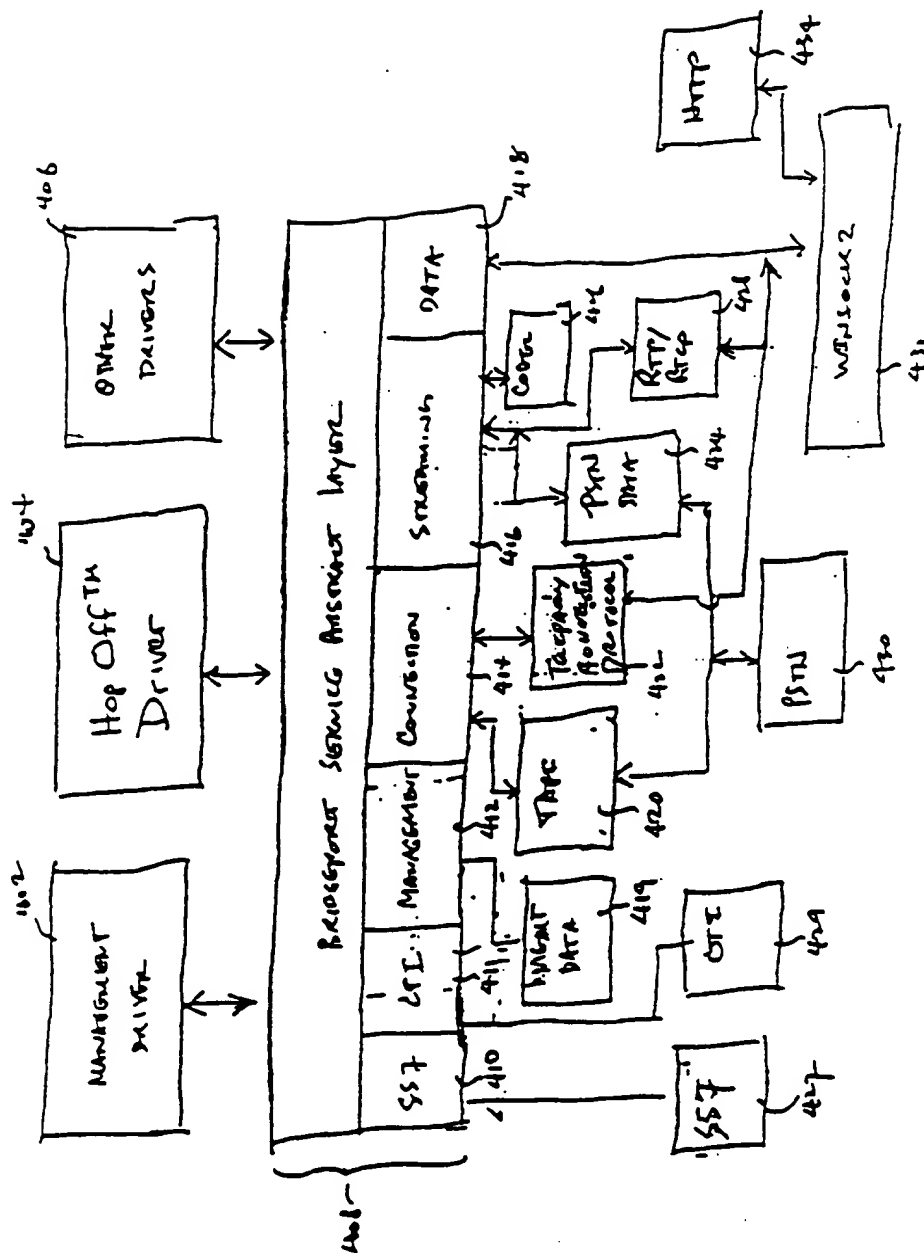


FIGURE 4

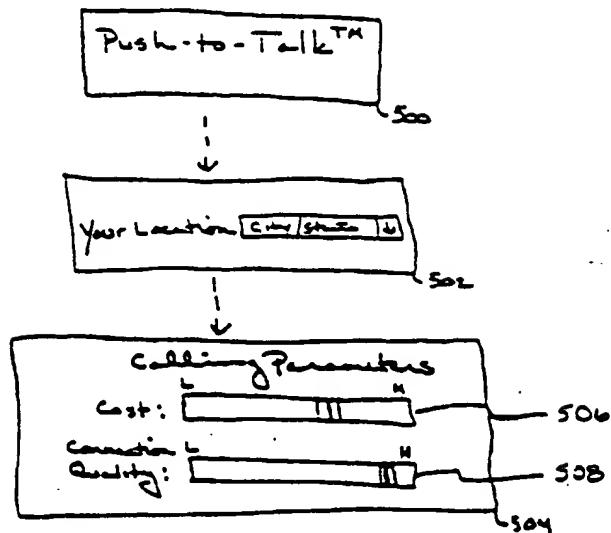
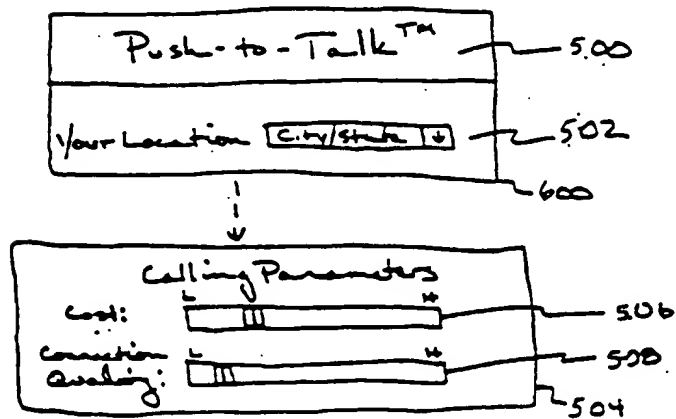
FIGURE 5FIGURE 6.

FIGURE 7

7.00

Push-to-Talk™ 500

Your Location: 502

Calling Parameters

Cost: 406

Connection 408

Quality: 504